IN THE SPECIFICATION

In the Specification, please replace the Abstract as follows:

A method and apparatus for enhanced Internet telephony ensures that communication between a source and destination [[in]] is not interrupted by common network address translation. According to one aspect of the invention, communication may continue through a router that employs network address translation. ~~Such continued communications prevent the router from closing the source's outbound port.~~

Please replace the first full paragraph on page 2 as follows:

A major drawback[[s]] of the above typical environment is the difficulty in accommodating the Network Address Translation (NAT) that is typically implemented by the router 30. As is commonly understood, a Dynamic Host ~~Computer~~ Configuration Protocol server running on the router 30 assigns private dynamic IP addresses to the Media Terminal Adapter 40 and computer 50; thus effecting Network Address Translation (NAT).

Please replace the first full paragraph on page 3 as follows:

Figure 2 is a schematic representation of an example environment that addresses the issue of router 30 losing an outbound port during an Internet telephone connection. In the Figure 2 example, at the Internet telephone service provider, a destination, e.g., a pre-proxy server 70, receives messages from the router 30. Pre-proxy server 70 records the private IP address of the Media Terminal Adapter 40 during, for example, the SIP registration process. It also records the network address translation communication port assigned by the router 30 to the Media

Terminal Adapter 40 to and from which it will send and receive messages, such as SIP messages. Upon registration, the Media Terminal Adapter 40 passes fields used to communicate with the pre-proxy server 70. Examples of fields that can be passed include, for example, the private IP address of the Media Terminal Adapter 40, the public IP address of the router 30, and port information. After the pre-proxy server 70 receives the information from the Media Terminal Adapter 40, the pre-proxy server 70 periodically sends, for example, blank UDP messages to the Media Terminal Adapter 40, which contain the same destination and source address as a typical SIP message would have. Other messages could be ~~use in stead~~ used instead of the UDP message. The message used should prompt the Media Terminal Adapter 40 to send a response to the pre-proxy server 70. The pre-proxy server 70 sends, for example, the UDP message to the router 30 using the public IP address of the router 30 and the port information received in the message from the router 30. The pre-proxy server 70 sends, for example, the UDP within the limited time that the router 30 maintains that private address assigned to the Media Terminal Adapter. The router 30 accordingly ~~routs~~ routes the message to the destination designated in the message from the pre-proxy server 70. The pre-proxy server 70 also maintains the private and public IP addresses of the MTA and rewrites the headers in the actual SIP messages based on this information.

Please replace the first full paragraph on page 5 after the heading "Description of the Preferred Embodiments" as follows:

Figure 3 is a schematic representation of an example environment employing an embodiment of the present invention. In Figure 3, a user initiates a call using a telephone handset 60. As described above, the media terminal adapter 40 implements standard signaling between itself and an Internet telephony regional data center 80. Once the user has been registered and the destination has provided a SIP acknowledgment of the SIP invite sent by the media terminal adapter 40, communication between the caller 60 and a customer in a destination area 100 proceeds using, for example, Real-time ~~Transfer~~ Transport Protocol (RTP) between the caller and [[the]] a customer in the destination area 100 via the Internet 20 and, for example, a RTP relay 90 in Internet telephony point-of-presence 110 in the destination area 100.

Please replace the first full paragraph on page 6 as follows:

However, with the call set up as described above, the router 30 may close the outbound port after a timeout period. As a result, voice data from the customer in the destination area 110 will not reach the telephone handset 60 behind router 30. To avoid the router 30 timing out and closing the outbound port, an embodiment of the present invention causes the media terminal adapter 40 to send an outbound message to the Internet telephony regional data center 80. One way of accomplishing this is to have the pre-proxy server 75 periodically send an empty SIP notify message to the media terminal adapter 40. The media terminal adapter 40 responds to this notify message in accordance with SIP standards by,

for example sending an acknowledgement message. The sending of a message by the media terminal adapter 40 causes the router 30 to keep the outbound port open by, for example restarting the router's timeout period.

Please replace the second full paragraph on page 6 as follows:

Referring to the exemplary embodiment shown in Figure 3, the Internet telephony regional data center 80 has the pre-proxy server 75 separated from the RTP relay 85. While this separation is not necessary to the present invention, in some environments, it [[is]] allows additional functionality to be more easily added to the pre-proxy server 75. An example of such additional functionality is the dynamic ~~allocate~~ allocation of the RTP relay 85. The pre-proxy server 75 can allocate the closest RTP relay between the two calling parties. That allocation enables the ability to decrease latency and travel time of the RTP stream. Also as shown in FIG. 3, with the exemplary embodiment, only SIP messages get routed to the Internet telephony regional data center 80. The RTP stream need not travel to the data center, and depending upon the location of the caller and the destination area 100, can travel within a limited geographic area. For ~~examle~~ example, the telephone handset could be located in California, and the Internet telephony regional data center 80 could be located in New Jersey. If the destination area 100 is also in ~~Calrfornia~~ California, the Internet telephony point of presence in the destination area 90 would be allocated by the pre-proxy server 75 to also be in California. Thus, as noted above, the RTP stream would remain in California; tending to reduce ~~to decrease~~ latency and travel time of the RTP stream.